A TIME-VARIANT REVERBERATION ALGORITHM FOR REVERBERATION ENHANCEMENT SYSTEMS

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ABSTRACT

This paper presents a new time-variant reverberation algorithm that can be used in reverberation enhancement systems. In these systems, acoustical feedback is always present and time variance can be used to obtain more gain before instability (GBI). The presented time-variant reverberation algorithm is analyzed and results of a practical GBI test are presented. The proposed reverberation algorithm has been used successfully with an electro-acoustically enhanced rehearsal room. This particular application is briefly overviewed and other possible applications are discussed.

1. INTRODUCTION

An electroacoustic enhancement system or a reverberation enhancement system (RES) is a system that is used to alter the sound field in a space using microphones, loudspeakers, and electronic circuits. Reverberation enhancement systems have been used in concert halls and multipurpose halls either to correct for inferior acoustic design or to provide means to change the acoustical properties of the hall. They are used in halls that don't naturally have enough reverberant energy. The theoretical aspects of reverberation enhancement system have been well explained by Svensson [1] and a review of commercial enhancement systems has been presented by Kleiner and Svensson [2].

The transfer functions present in all electroacoustic reverberation enhancement systems are depicted in Fig. 1. The actual transfer functions vary depending on the implementation of the system and the acoustics of the room. The acoustic feedback $H_{LM}(\omega)G_{ML}(\omega)$ is a major problem in these systems. Transducers with non-flat frequency response, positioning of the microphones (usually far from sound sources for practical reasons), and the reverberation algorithm make things even worse.

A multi-channel RES in a room forms a multi-feedback system. There is a limit to the open-loop gain, above which there is an uncontrollable positive feedback between the microphones and the loudspeakers. The maximum gain before the system becomes unstable is called gain before instability (GBI). Close to instability, the frequencies, at which the feedback loop transfer function has its maximum values, become audible as changes in timbre or as ringing tones. This coloration may be discernible already at levels 12 dB before the system becomes unstable. Typically, the peaks in transfer function are on average approximately 10 dB higher than the mean magnitude. This means that the system becomes unstable already at levels that are 10 dB lower than the wide-band average gain would suggest [1].

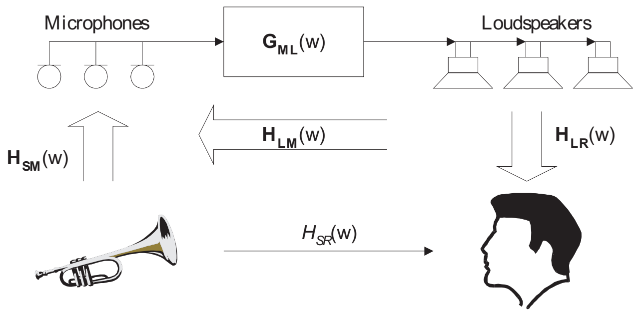

Figure 1: *The transfer functions present in all RESs, after [1].*

The coloration can be reduced using equalization. However, the loop transfer function still has sharp peaks that are heard as ringing tones and indicate possible instability. An efficient way to decrease the feedback problem is to use time-varying algorithms in $G_{ML}(\omega)$. With time-varying algorithms, the loop transfer function changes continuously, preventing the rise of self-generating sustained peaks. Time variance can be implemented as amplitude, delay, or phase modulation or as frequency shifting. The most efficient modulation method is phase modulation [3], but it has been claimed hard to implement. If the system transfer function $G_{ML}(\omega)$ contains an additional reverberation algorithm, the time variance can also be implemented directly in the algorithm. In this paper, we present a time-variant version of a known reverberation algorithm [4] that is a special case of the feedback delay network algorithm [5].

2. THE PROPOSED TIME-VARIANT REVERBERATION ALGORITHM

A reverberation enhancement system (Fig. 1) can operate on the whole audio range or on a narrow band. There are also systems with no additional reverberation algorithm in $G_{ML}(\omega)$. The choice of bandwidth and number of channels is application dependent. In this study we concentrate on a system that is a wideband multichannel system containing additional reverberation. The block diagram of the applied reverberation algorithm is depicted in Fig. 2.

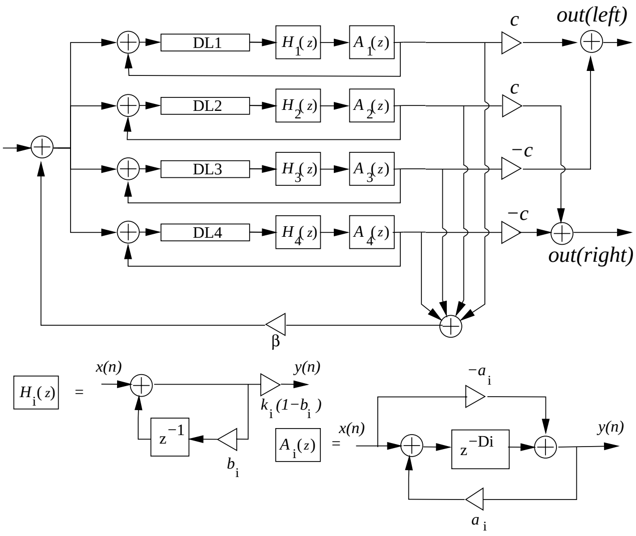

Figure 2: *The applied reverberation algorithm, containing four channels [4].*

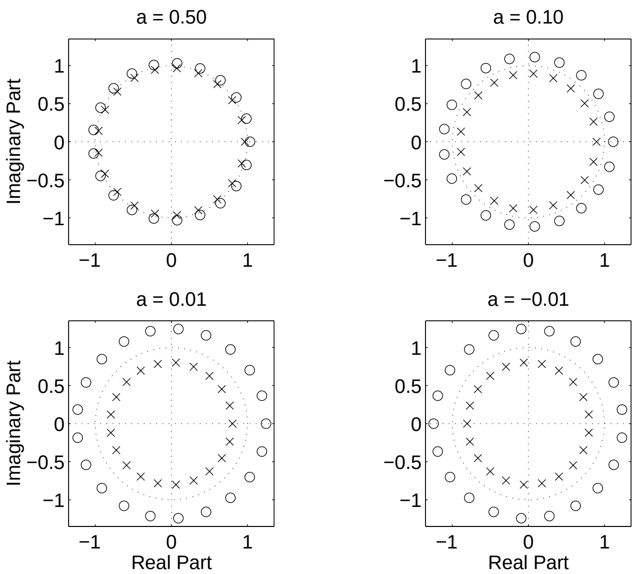

Figure 3: *The pole-zero plot of one comb-allpass filter with different a_i values. The delay z^{-Di} inside the filter is 21 samples (defines the number of pole-zeros pairs).*

Each channel of the reverberation algorithm contains a delay line, a lowpass filter, and a comb-allpass filter. In addition, the channel is fed back to itself and to other channels (see Fig. 2). This algorithm produces natural sounding diffuse late reverberation and it has been used successfully in the DIVA auralization system [6].

The reverberation algorithm is made time-varying by modulating the feedback coefficient a_i in the comb-allpass filter $A_i(z)$ on each channel. The delay line z^{-Di} in such a comb-allpass filter is several hundred samples long. The modulating function can be any continuous signal; e.g., a sinusoidal signal with frequency of a few Hertz has been found practical.

2.1. Analysis of the time-variant comb-allpass filter

The transfer function of the comb-allpass filter is

$$A_i(z) = \frac{-a_i + z^{-N}}{1 - a_i z^{-N}}, \tag{1}$$

where a_i is the feedback coefficient, and N ($= Di$ in Fig. 2) is the length of the delay inside the filter. The pole locations of this filter are given by

$$p_n = Re^{2n\pi/N}, \quad n = 0, \ldots, N-1 \tag{2}$$

for negative a_i, and by

$$p_n = Re^{(2n+1)\pi/N}, \quad n = 0, \ldots, N-1 \tag{3}$$

for positive values of a_i. In the above equations, R is the distance of the poles from zero. It is given by

$$R = \sqrt[N]{|a_i|}. \tag{4}$$

The filter is an allpass filter with the zeros of the transfer function being reciprocals of the poles. The arrangement of poles and zeros is depicted in Fig. 3 for different values of a_i and $N = 21$.

As the value of a_i is changed, say, from 0.9 to -0.9, the poles move from close to the unit circle across the zero to again near the unit circle. The comb-allpass filter works by introducing a frequency dependent delay to the signal. The closer to the unit circle the poles are, the greater is the effective length of the impulse response of the filter, i.e., the more it is going to delay certain frequencies.

Figure 4 shows the time domain effect of modulation on one channel with different values of a_i. It can be noted that the instantaneous impulse response changes quite a lot with the value of a_i. It can also be seen that the reflection density changes as a function of a_i. On average, however, the reflection density of the whole reverberator is rather constant. In frequency domain, the effect of modulation is seen in Figs. 5 and 6. As expected, the center frequencies of the peaks are shifted with different a_i values and this reduces the feedback efficiently. The shift of the frequencies is due to the changes in group delay caused by the modulated comb-allpass filter.

2.2. Discussion

The modulation is not easily perceivable if the reverberator contains at least four channels and the modulation frequencies and phases differ from channel to channel. The modulation shifts frequency peaks in different directions depending on the signal frequency, so no perceivable pitch changes occur due to the modulation.

The frequency dependent delays introduced by the comb-allpass filters, change the reverberation time of the whole reverberator at certain frequencies. However, the effect on the overall reverberation time is small enough to be left without further consideration. The delays may become significant, if very short reverberation times are required. This is usually not the case with reverberation enhancement systems.

One of the drawbacks of the current modulation method is

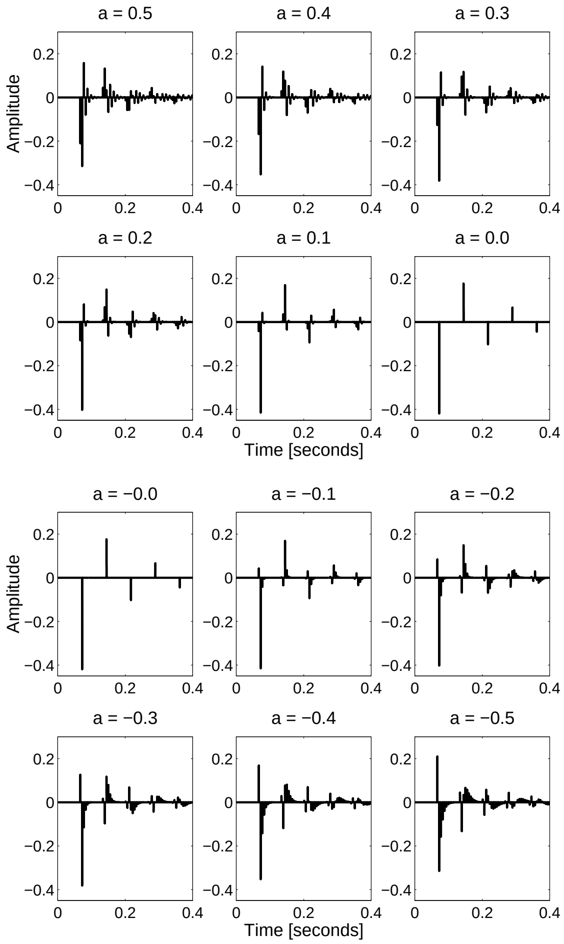

Figure 4: *Impulse responses of one channel at different a_i values.*

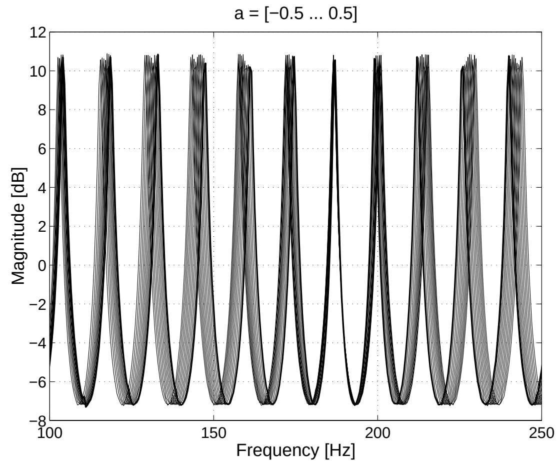

Figure 5: *Part of the frequency response of one channel at different a_i values.*

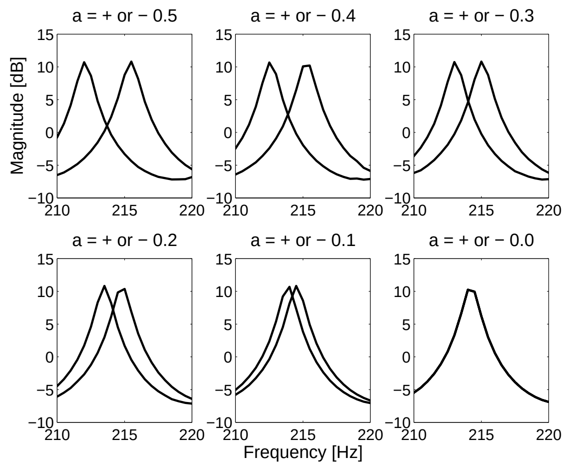

Figure 6: *The shift of one frequency peak in one channel at different a_i values.*

that the delays inside the comb-allpass filters must be selected carefully; otherwise there will be some frequencies that are only slightly affected by the modulation. This can be seen also in Fig. 5, where one of the peaks (at approx. 186 Hz) is only slightly modulated.

3. EXPERIMENTAL TEST OF THE BENEFITS OF MODULATION

To get an idea about the effect of the modulation to GBI we did a practical test in an office meeting room. The room dimensions were 4.5 m x 7.0 m x 2.6 m and the volume was 82 m^3. The reverberation time in the room without the system was 0.5 s at low frequencies and 0.4 s at mid and high frequencies.

3.1. System configuration

The test system consisted of a SGI O2 workstation, a multichannel D/A converter, four active loudspeakers (frequency response 70 - 20000 Hz), and an omni-directional microphone with a custom pre-amplifier. The reverberation algorithm was implemented using C++ and run in real time. Naturally, the system hardware introduced some delay from input to output due to hardware and software buffering in the SGI. We measured this delay with a DAT recorder. With the buffering and the hardware setup used in the test the delay was 25.1 ms.

The reverberation algorithm contained eight channels, two to each loudspeaker. This way each loudspeaker reproduced incoherent signal compared to other loudspeakers. The delay line lengths (DLi, see Fig 2) were between 2957 and 4099 samples (corresponding to 67.1 and 92.9 ms, respectively). The lowpass filters $H_i(z)$ after each delay line implement the frequency dependent reverberation time. The filter coefficients were calculated according to the ideas and formulas presented by Moorer [7] and Jot and Chaigne [5]. Finally, the comb-allpass filters $A_i(z)$ had delay line lengths between 233 and 443 samples (corresponding to 5.3 and 10.0 ms). The modulation functions that modulated coefficients a_i, were sinusoids at 2.6 - 3.5 Hz (different frequency for each channel).

An important remark about the system is that it produced only late reverberation, i.e., there was no attempt to generate additional

RT 1.0	**R1**	**R2**	**R3**	**Average**
No mod.	-11.1	-10.3	-8.1	**-9.8 dB**
$a_i \Updownarrow \pm$ **0.5**	-8.6	-9.0	-5.0	**-7.5 dB**
$a_i \Updownarrow \pm$ **0.8**	-6.4	-7.3	-4.0	**-5.9 dB**
$a_i \Updownarrow \pm$ **0.9**	-5.4	-5.9	-2.9	**-4.7 dB**

RT 1.5	**R1**	**R2**	**R3**	**Average**
No mod.	-14.1	-13.5	-10.5	**-12.7 dB**
$a_i \Updownarrow \pm$ **0.5**	-11.7	-11.9	-7.5	**-10.4 dB**
$a_i \Updownarrow \pm$ **0.8**	-9.0	-9.7	-5.9	**-8.2 dB**
$a_i \Updownarrow \pm$ **0.9**	-7.6	-8.2	-4.9	**-6.9 dB**

RT 1.8	**R1**	**R2**	**R3**	**Average**
No mod.	-14.6	-14.8	-12.4	**-13.9 dB**
$a_i \Updownarrow \pm$ **0.5**	-12.9	-12.8	-9.0	**-11.6 dB**
$a_i \Updownarrow \pm$ **0.8**	-9.8	-10.2	-7.6	**-9.2 dB**
$a_i \Updownarrow \pm$ **0.9**	-8.6	-9.9	-5.9	**-8.1 dB**

Table 1: GBI values with different modulation gains when reverberation time was 1.0, 1.5, and 1.8 seconds.

	RT 1.0	**RT 1.5**	**RT 1.8**	**Average**
$a_i \Updownarrow \pm$ **0.5**	2.3	2.3	2.3	**2.3 dB**
$a_i \Updownarrow \pm$ **0.8**	3.9	4.5	4.7	**4.4 dB**
$a_i \Updownarrow \pm$ **0.9**	5.1	5.8	5.8	**5.6 dB**

Table 2: Total enhancement to GBI (no modulation vs. modulation), numbers collected from tables 1-3.

early reflections. The first output from the system is 92.2 ms after the sound is capture by the microphone. This initial delay contains the workstation delay (25.1 ms) and the shortest delay line length (67.1 ms).

3.2. Test procedure

To find out the effect of time variance to the GBI we measured the GBI values of a time-invariant version ($a_i = 0.5$) and several different time-varying versions. The GBI was defined as follows; it is the input gain with which the system becomes unstable. Instability occurs at a frequency dependent on the room and the system. For initial excitation, we used impulse-like sounds from hand claps to heavy books falling on a table or on the floor. We carried out the GBI test with three microphone positions and in each of them we repeated the test four times.

The reverberation algorithm contains dozens of parameters that can be varied. Such parameters are the number of channels, the length of the delay lines (also inside comb-allpass filters), the desired reverberation time, the modulation depth, etc. In this test we just fixed parameters without knowledge about the optimum values. However, the parameters used provided natural sounding reverberation and were suitable for our test because we were looking for the differences of time-invariant and time-variant versions. Finally, with time-variant versions we varied two parameters, namely the reverberation time of the reverberator algorithm (RT, three values 1.0, 1.5, and 1.8 s) and the modulation depth of the coefficient a_i (from -0.5 to 0.5, from -0.8 to 0.8, and from -0.9 to 0.9).

3.3. Results

The results of the obtained GBI values are collected to tables 1 and 2. Each number in table 1 is the average of four measurement. The acronyms R1, R2, and R3 denote different receiver points.

The results show that GBI values are dependent on the measuring point. This is natural, because in each position the room has a different mode distribution, which affects the instability (different peaks in the frequency response are amplified). The GBI values are also a function of the reverberation time and the amplitude of modulation, as expected. The more reverberation is added to the system the lower GBI values occur.

The results confirm that with the time-varying version we can obtain more gain before instability than with the time-invariant reverberation. Depending on the amplitude of the modulation we could raise the gain between 2 and 6 dB comparing to the time-invariant version (see Table 2). This is an important improvement and is in line with the results of Nielsen and Svensson [3].

3.4. Discussion

The time variance becomes more and more audible when the amplitude of the modulation is raised. The modulation is especially audible near the GBI; then the modulation is heard as changes in coloration. However, in real use the gain should be so low that coloration is not heard as sustained tones. On the other hand, higher GBI is obtained when the modulation amplitude is high. The compromise between optimum prevention of the feedback and the audibility of the modulation should be studied with listening tests.

The efficiency of the presented modulation at different frequencies is still a subject to be explored. Especially the effect on low frequency feedback needs to be studied. In our test system, the loudspeakers had a limited low frequency response, and the room did not have strong resonances either, so low-frequency feedback was not a problem in the test system.

4. APPLICATIONS

A time-variant reverberation algorithm has been used successfully with a prototype of an electro-acoustically enhanced rehearsal room [8]. The idea in such a room is that the room is a copy of the stage of a concert hall. Instead of the audience area, there is an anechoic wall which absorbs all sound energy (see Fig. 7). The anechoic wall is equipped with a reverberation enhancement system that produces the reverberant response that usually returns from the hall to the stage. The benefits of such a system are that it requires less space than the whole concert hall, the subjectively important early reflections correspond the early reflections of a real stage, it sounds (almost) like a real concert hall, the sound pressure level in such a room is reasonable, and the reverberation time of the room can be changed. Briefly, such a rehearsal room is a space and cost effective solution for a symphony orchestra that can not afford to have all rehearsals in a real concert hall.

The prototype of this kind of rehearsal room was constructed in a big music studio, with dimensions 20 m x 15 m x 7 m. One wall was covered with several layers of Molton curtains to make it as anechoic as practically possible for the prototype. In this system we used 24 small active loudspeakers and 4 subwoofers to

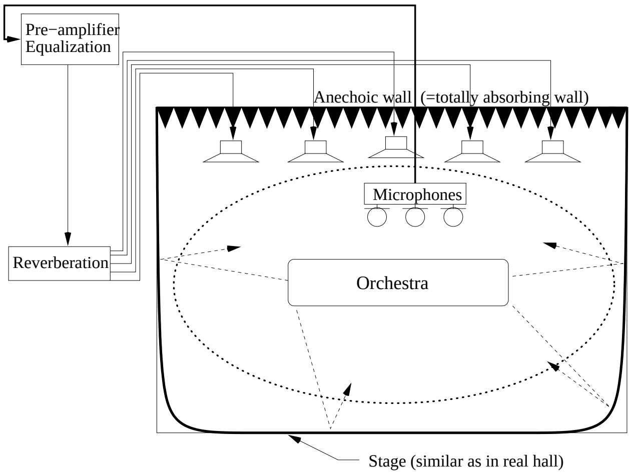

Figure 7: *The example application: Rehearsal room for symphony orchestra [8].*

reproduce the late reverberation that was implemented with the time-varying algorithm presented in this paper. The prototype was evaluated subjectively by a group of musicians who played in the room. Their comments were promising and they claimed that the room really sounded like a real concert hall.

Another situation where such a "virtual concert hall" can be applied is in outdoor concerts of symphony orchestras. In a way similar to the rehearsal room, we can construct the stage to an outdoor venue (e.g. a park), where the anechoic wall is not needed (the sound propagates to free space from the stage). To enhance the playing conditions of the musicians, small loudspeakers can be mounted between the orchestra and the audience without overly disturbing the visual impression. This way the orchestra is provided with an enhanced and natural sounding monitoring and a normal public address system can be used for the audience.

The presented time-varying reverberation can also be used in several other active acoustic scenarios. One such scenario is a music practicing system at home. Domestic multichannel sound reproduction is more and more common because of the 5.1 sound systems in our living rooms. By adding a microphone and a late reverberation algorithm to a 5.1 sound system one can play and practice his/her instrument in various acoustical conditions. The time-variance prevents the acoustical feedback which usually is problem in such a system. Another scenario is to use time-variant reverberation in virtual environments, such as CAVEs [9]. The realism of the virtual environment can be improved by adding a realistic soundscape. The user of the virtual environment can have a microphone and all the sound he/she makes is processed with suitable reverberation that corresponds to the visual model of the virtual space. The plausibility of the virtual auditory environment can be enhanced by rendering also early reflections with the reverberation [10].

5. CONCLUSIONS

In this paper, we have presented a time-variant reverberation algorithm for reverberation enhancement systems. The time variance makes reverberation more robust to the feedback problems that are always present in reverberation enhancement systems. The influence of the time variance was analyzed and the results of a practical test were reported. The test showed that with time variance we can obtain from 2 to 6 dB (depending on the setup of the system) more gain before instability compared to the time-invariant version of the same reverberation algorithm. Despite the fact that the algorithm contains some inherent weaknesses, the results are promising. However, more thorough testing is needed to answer the open questions about the audibility of the modulation, and the effect of the modulation at different frequency bands. The proposed algorithm can be used in several applications, of which we have in this paper described a few.

6. ACKNOWLEDGMENTS

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